Release Notes
v6.11.1 for 96x Multi-Channel Test Unit

This document describes a new software release (v6.11.1) for the 96x. This software supersedes Version 6.4.12 and is available now.

NEW FEATURES

1. **Real-time PESQ** — PESQ (Perceptual Evaluation of Speech Quality) is another widely used method (per ITU-T Recommendation ITU-T P.862) of evaluating voice quality of VoIP (voice over IP) calls. In addition to a MOS (mean opinion score) number, it also provides: PESQ score, PESQ Ie (equipment value) and PESQ Lq (listening effort number).

   Two operating modes are available:
   a. Automatic Director and Responder mode that uses inband telemetry
   b. Manual transmitter and receiver mode that performs one-way measurement only without inband telemetry

2. **Remote Audio** — This feature enables a remote user to audibly monitor test progress at the 960B location. It also allows the remote user to talk over the circuit under test. This feature is now available for the 960B PCM and Analog test interfaces.

3. **WAV File Sender for Ethernet and Analog Interfaces** — (This feature is already available for PCM interfaces). Each channel can download a maximum length of 32-second worth of PCM (mu or A-law) data. When playing, the downloaded data will be continuously repeated. This capability is available in the Send Tone and Echo Generator test functions.

4. **Capture of Audio Signals in PCM or WAV Format** — Users are now given a choice of capturing circuit audio in PCM or WAV format. Previously, only PCM format was supported.

5. **Manual Mode for Testing Over Ethernet Interfaces** — Allows a user to change VF tests dynamically while the SIP call stays connected.

6. **Digit Sender for Tests Over Ethernet Interfaces** — Allows a user to dynamically interrupt a test to send DTMF/MF digits for credit card calls, voice conference bridge calls, etc.

7. **RTP Loader Test** — Useful for loading IP networks, this function supports multiple end points simultaneously generating true RTP packets with G.711 encoding.

   For 96x units equipped with the new "IP2" Ethernet board, this function supports up to 400 end points simultaneously generating true RTP packets with G.711 encoding. Therefore, a user can generate up to 200 SIP/RTP calls from one 960B Ethernet interface to the other, or a combined total of 400 calls from both interfaces to external end points. These quantities are based on configuring the 960B Ethernet interfaces for full duplex 100BaseT. Lower speed configurations will reduce the quantity of simultaneous call supported.

   Note that units equipped with the older "IP" board are limited to a maximum of 64 end points per full duplex 100BaseT circuit.

8. **Auto Configure of IP Channel IP MAC Addresses** — When using the Multi-Cfg (multiple configuration) button to configure multiple channels at once, you can now have the 96x GUI

automatically provision a unique, IP address, MAC address, and or "phone number" for each of the selected channels. The function is accomplished by use of the new "Auto" buttons for each parameter.

9. **T.38 FAX Monitor** — The IP T38 FAX monitor function works like RTP monitor. One can specify a T38 monitor under each virtual channel, and each virtual channel will capture a specific FAX call pair. All T38 raw packets will be decoded and interpretations are provided for the V21 control messages. This feature only captures T38 packets transported using UDPTL (Facsimile UDP Transport Layer protocol), as exactly specified in ITU T.38.

10. **PDD (Post-Dial-Delay) for SIP and DS1**
    Previously, the 960B only measured PDD in analog mode (defined as the time between the end of the last digit and beginning of the ringing back signal). The PDD measurement function has now been broadened to the SIP call and DS1 interface.

    a. Under SIP call, PDD is defined as the time between sending the "INVITE" message and receiving the "180 Ringing" message.

    b. In DS1 CAS mode, PDD is defined the same way as in analog (the time between the end of the last digit and beginning of the ringing back signal).

    c. In ISDN mode, PDD is defined as the time between sending "Setup" message and receiving "Call Alert" message back.

11. **Command Line Interface** (CLI)
    a. **"unit" command**
       1) Now supports an **ip** parameter to allow an *admin* user to change the Ethernet remote control port's IP address

       2) Now reports 96x memory utilization: Total Memory and Available Memory, in bytes

    b. **Sip Monitor** — added the SIP Monitor test function "sipmon"

    c. **Dual Spectrum Analyzer** — added the Dual Spectrum Analyzer test function "dualspec"

    d. **RFC 2833 Monitor** — added the RFC 2833 Monitor test function "rfc2833"

    e. **Logging SIP Call Information** — Previously, if the -log flag is added to a test (such as smosd) the results of the VF test were logged but the SIP Call information was not. There is now an xx_ip.csv file generated for IP tests whenever a VF test log is requested

    f. **Digit Receiver** (digrecv) — Added command line parameters to specify allowed min/max criteria for digit reporting (defaults are shown within curly brackets {}):

       [-minon]   Minimum On Duration (ms) {40}(Optional) 30 to 100
       [-minlvl] Minimum Tone Level (dBm) {-25}(Optional) -35 to 5
       [-maxtwist]Maximum Tone Difference (dB) {6}(Optional) 0 to 10
       [-maxdf]   Maximum Frequency Deviation (Hz) {10}(Optional) 0 to 50
       and
       [-dur]   Test Duration (sec) {0(forever)}(Optional) 0 to 1000

    g. **Digit Sender** (digsend) — Added IP-related command line parameters (defaults are shown within curly brackets {}):
       [-localip]   (IP only) Local IP Address (Optional)
       [-localport] (IP only) Local IP Port {5060} (Optional) 1024 to 65535
       [-localmac] (IP only) Local MAC Address (Optional)

[-localphone] (IP only) Local Phone Number (Optional)
[-jitter]   (IP only) SIP Jitter Buffer Size {40} (Optional) 10 to 100
[-siptos]   (IP only) SIP ToS (8 binary digits) {00000000} (Optional)
[-rtptos]   (IP only) RTP ToS (8 binary digits) {00000000} (Optional)
[-port]   (IP only) Destination IP Port {5060} (Optional) 1024 to 65535
[-phone]   (IP only) Phone Number (max 32 chars) {} (Optional)
[-pktsize]   (IP only) Packet Size (ms) {20} (Optional) 10 to 40
[-decoder]   (IP only) Decoder (PCMu or PCMa) {PCMu} (Optional)
[-user]   (IP only) Authentication User Name (max 32 chars) {} (Optional)
[-pass]   (IP only) Authentication Password (max 32 chars) {} (Optional)
[-regip]   (IP only) Registration IP Address (Optional)
[-regport]   (IP only) Registration IP Port {5060} (Optional) 1024 to 65535
[-regexp]   (IP only) Registration Expiration (min) {10} (Optional) 1 to 255
[-regint]   (IP only) Registration Interval (sec) {30} (Optional) 1 to 50

h. **Echo Generator** (echogen)
   1) Added the "-dur" parameter, enabling users to specify the test duration

   2) Added the "-wav" parameter, enabling users to mix a WAV file with the echo it generates

   3) Added the "-encoder" parameter for Analog and IP test interfaces. This parameter allows users to identify whether PCM audio files are mu-Law (µ-Law) or A-Law encoded, for proper decoding before transmission.

i. **Send/Measure Tone** (smtone)
   1) Added noise options to the smtone test function. Users can now choose to send Tone, White Noise, Hoth Noise, ERL noise, SRL-LO noise, or SRL-HI noise.

   2) Added WAV/PCM play option to the smtone test function. Users can now choose to simultaneously transmit a WAV file while sending tone or noise.

   3) Added the "-encoder" parameter for Analog and IP test interfaces. This parameter allows users to identify whether PCM audio files are mu-Law (µ-Law) or A-Law encoded, for proper decoding before transmission.

j. **PVIT** (pvit) — Added command line parameters to allow individual Tx and Rx control to the "pvit" test function:

   [-txonly] Transmit only(Optional)
   [-rxonly] Receive only(Optional)

k. **SMOS Responder** (smosr) — Added command line parameter to allow user adjustment of the SMOS Responder timeout timer (default is shown within curly brackets {}):

   [-tptdur] TPT Retry Duration (sec) {0}(Optional) 0 to 60

l. **FAX Transmit Test** (faxtx) — Added command line parameter to allow user adjustment of the number of FAX test pages to transmit (default is shown within curly brackets {}):

   [-pages]   Pages to send {2} (Optional) 2 to 100

m. **Echo Ping Server** (echoping) — Added the ICMP Echo Server (ping) test, for IP test access interfaces ONLY. This function enables an Ethernet test access port to automatically answer far end "ping" requests.

> echoping -if [-rn]

>   -if   IF# 1 to 6
> [-rn]   Resource# (Optional) 1 to 255
> No spaces or tabs allowed in parameters.

n. **Remote Audio** (remaudio) — Added the "remaudio" command for establishing remote audio call connections (default parameter values are shown within curly brackets {}).

> remaudio [-dn] [-sn] -if [-rn] [-log] [-dir] [-resp] [-mf] [-monif] [-monrn] [-monmode]

> [-dn]   destination(Optional) Characters without whitespace
> [-sn]   source #(Optional) Characters without whitespace
>   -if   IF# 1 to 10
> [-rn]   Resource#(Optional) 1 to 255
> [-log]   Log File(Optional) Characters without whitespace
> [-dir]   Be Director: # times to run {0}(Optional) 0 to 999
> [-resp]   Be Responder(Optional)
> [-mf]   Send MF call setup digits (CAS only) {DTMF}(Optional)
> [-monif] Interface to Monitor {1}(Optional) 1 to 10
> [-monrn] Resource to Monitor {1}(Optional) 1 to 99
> [-monmode] Monitor Mode (off, tx, rx, or both) {both}(Optional) Characters without
>   whitespace

o. **PESQ** — Added real-time PESQ (Perceptual Evaluation of Speech Quality) commands:
> pesqrx   PESQ Receive Test
> pesqtx   PESQ Transmit Test
> spesqd   SPESQ Director Test
> spesqr   SPESQ Responder Test

p. **ISDN Span Call Setup Parameters** — Added ISDN "Number Type" and "Number ID" call setup configuration parameters.
1) **"Number Type"** choices are: Unknown, International (E1 default), National (T1 default), Network Specific, Subscriber, and Abbreviated.

2) **"Number ID"** choices are: Unknown, ISDN Plan (default), Data Plan, Telex Plan, National Plan, and Private Plan.

ENHANCEMENTS
1. **SIP Calls**
   a. The SIP call engine now handles cases where incoming SIP message headers contain varying upper or lower-case characters for the key words

   b. Now handle short-form caller-ID data

   c. For the Via-header line, we now accept a line length up to 750 characters (used to be 250).

   d. The 960B now allows user specification of RTP jitter buffer size down to ms resolution (previously was packet-size resolution).

e. All delay-sensitive measurements (SMOS, PESQ, Echo Sounder, Echo Generator and One-way-delay) are robust and contain no uncertainties.

f. Local identity for each SIP virtual channel: Under each SIP director or responder virtual channel, one can now also specify a unique set local addresses that include (1) SIP source phone number, (2) SIP source port number, (3) SIP source IP address and (4) SIP source MAC address.

g. SIP registration is now enabled in both Director and Responder modes

h. The user-name string has been added to the "Contact" header line. Some registration servers are not happy when the user name string is missing.

i. Also changed the "pace" of DHCP discover requests. They have been slowed from 2 seconds to 5 seconds wait time before issuing another DHCP-discover request. It still tries 7 times and before timing out.

j. Expanded list of SIP error messages (total of 62) for more complete identification of SIP call failures.

2. **SIP Monitor**
   a. **Support for Virtual-LAN-formatted Ethernet frames** — This allows users to RTP-monitor an enterprise LAN network where VLAN formats might be used. No special configuration is needed. The DSP auto-detects the VLAN frames. The DSP only receives VLAN frames; it does not send VLAN frames. TX ability will be officially added in the next release phase.

3. **FAX Test**
   a. **Multi-Page Fax Test** — A user can now specify sending 2 to 100 pages of test pattern. All pages after page 2 contain the same pattern as page 2.

   b. **If initial 2100Hz CED tone is missing**, but the ensuing DIS FSK signal shows up, the FAX test will still continue but will report "CED tone missing"

   c. The **ON and OFF time-stamps** of each FAX signal event (the initial tones, protocol control messages, training signals, and page transmissions) are now provided

4. **RTP Monitor**
   a. **Now tolerates asymmetry** — When a pair of RTP streams contain different codec types or packet sizes for each direction, the new RTP monitor can now handle them (The old RTP monitor would claim finding only one side).

   b. **Support for Virtual-LAN-formatted Ethernet frames** — This allows users to RTP-monitor an enterprise LAN network where VLAN formats might be used. No special configuration is needed. The DSP auto-detects the VLAN frames. The DSP only receives VLAN frames; it does not send VLAN frames. TX ability will be officially added in the next release phase.

5. **G.168 Test**
   a. For serial port control of echo cancellers, the allowable Com Port number has been increased from 4 to 8, to accommodate USB-serial adaptors often used with laptop computers

6. **Send/Measure Tone** — You can now set the duration of the Send/Measure Tone test when using the analog interface

7. **GUI** (graphical user interface)
   a. Now **remembers the IP address of the last 5 remote units**, noticeably speeding up connections to remote units.

b. **Added Reference Signal Status in One Way Delay** — Some users may not know they need to provide a reference signal when running the One Way Delay test. Displaying the reference signal status reminds them.

c. **Minimum Delay Between Tests** — In the test setup screen, the "Load Type" window allows a user to specify the minimum delay between repeating tests. Previously, the delay units were in milliseconds, and allowed a maximum delay of a little over 16 minutes. The units are now in seconds, and the maximum allowed value is now 999,999 (277 hours).

CORRECTIONS

1. **G168 Test**
   a. The echo canceller serial communication log now contains reply information from the echo canceller

2. **Command Line Interface** (CLI)
   a. The **Ctrl-C** key sequence — can now be used to break out of blocking commands. Previously, Ctrl-C would terminate the remote control session.

3. **PCM Status LEDs** — Front panel status LEDs now change from Green to Red when an ISDN D-Link is in an error state

4. **CLI Memory Leak Problem** — Corrected a memory leak problem in the Command Line Interface which could cause a system crash after several days of heavy use

5. **MOIP Director Log** — Corrected a problem where the results log file created by the MOIP test put non-numeric text into the same cell as numbers. This prevented users from performing analytical analysis of the results in a spreadsheet program.

   Examples of this problem: Gain columns had a number followed by dB, Delay had number followed by ms, Percent columns had a number followed by %, etc.

KNOWN DEFECTS

1. **RFC2833 Monitor** — The reported PT (Payload Type) number is incorrect. This is fixed in unreleased "Engineering Test" IP DSP files v63.4 and higher. Contact Sage Technical Support if you require this minimally tested patch.

2. **ICMP_ECHO** (ping response) — This "test", when run, should answer Ethernet "ping" requests. This function was inadvertently broken during development of this release. This is fixed in unreleased "Engineering Test" IP DSP files v63.3 and higher. Contact Sage Technical Support if you require this minimally tested patch.

3. **SMOS & PESQ** — Under poor transmission conditions, the telemetry signals can be corrupted sufficiently to cause erroneous results to be reported without declaring a test failure. This is fixed in the following unreleased "Engineering Test" DSP files (or later): t1.cof v74.4, ip.cof v63.3, and pots.cof v40.3. Contact Sage Technical Support if you require these minimally tested patches.

   The patches add additional error checking. When transmission errors are detected, the test will enter the SMOS IDLE state and send a loud NACK (negative acknowledgment) tone to the other end for 3 seconds, then declare an SMOS ERROR state.

4. **One Way Delay** — This test can give inaccurate results in DS1 mode when both spans are sending and measuring in different channels. This is fixed in unreleased "Engineering Test" T1 DSP files v74.4 or later. Contact Sage Technical Support if you require this minimally tested patch.

Note that, with this patch, there may still be buffer uncertainty (7 or 8ms, or 504 or 505 ms). This can be eliminated by going to the span configuration screen and clicking the "SAVE" button.

5. **ISDN D-Channel Call Setup** — When the ISDN number type is set to "Abbreviated", the call setup message properly skips the octet "0xa1" which means "Send Complete". However, this has caused some interoperability problems with some switching machines. This is fixed in unreleased "Engineering Test" T1 DSP files v74.4 and higher. Contact Sage Technical Support if you require this minimally tested patch.